Claim Amendments

Please amend claims 21-25 as follows:

1. (original) A method for adaptively controlling level of a receiver buffer in a client in a multimedia streaming network, the streaming network comprising a server for providing streaming data to the client, wherein the receiver buffer is used to compensate for difference between data transmission amount by the server and data usage amount by the client so as to allow the client to have sufficient amount of streaming data to play-out in a non-disruptive manner, said method comprising:

defining in the client at least one parameter for determining a rate adaptation operating range so as to carry out rate adaptation between the server and the client;

adapting in the server the data amount to a reception rate based on said at least one parameter; and

adjusting in the client packet transfer delay variation based on said adapting.

2. (original) The method of claim 1, wherein said at least one parameter comprises a minimum shift amount indicative of a difference between a sampling time and a transmission time of a packet at the server so as to allow the server to carry out said adapting based on the minimum shift amount.

3. (original) The method of claim 1, wherein said at least one parameter comprises a target shift amount indicative of a shift amount greater than a difference between a sampling time and a transmission time of a packet at the server so as to allow the server to carry out said adapting based on the target shift amount.

4. (original) The method of claim 1, wherein said at least one parameter comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

5. (original) The method of claim 1, further comprising the step of adapting a sampling rate to the transmission rate in the server based on said at least one parameter.

6. (original) The method of claim 1, wherein said at least one parameter comprises a clock shift amount for preventing playout disruption in the client.

7. (original) The method of claim 1, wherein said adapting comprises an adjustment of a transmission rate.

8. (original) The method of claim 1, wherein said adapting comprises an adjustment of a sampling rate.

9. (original) The method of claim 1, wherein said adapting comprises an adjustment of both a transmission rate and a sampling rate.

10. (original) The method of claim 1, wherein said at least one parameter comprises:

a minimum shift amount indicative of a difference between a sampling time and a transmission time of a packet at the server;

a target shift amount indicative of a shift amount greater than a difference between a sampling time and a transmission time of a packet at the server;

a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled; and

a clock shift amount, and wherein two or more of the minimum shift amount, the target shift amount, the specifying number and the clock are sent together to the server.

(original) A multimedia streaming system comprising:

at least a client; and

a server for providing streaming data to the client, the client having a receiver buffer to compensate for a difference between data transmission amount by the server and data usage amount by the client so as to allow the client to have sufficient amount of streaming data to play-out in a non-disruptive manner, wherein the client comprises:

a mechanism for defining at least one parameter for determining a rate adaptation operating range so as to allow the server to adapt the data amount to a reception rate based on said at least one parameter; and

a mechanism to adjust a packet transfer delay variation based on said adapting.

12. (original) The multimedia streaming system of claim 11, wherein said at least one parameter comprises a minimum shift amount indicative of a difference between a sampling time and a transmission time of a packet at the server so as to allow at the server to carry out said adapting.

13. (original) The multimedia streaming system of claim 11, wherein said at least one parameter comprises a target shift amount indicative of a shift amount greater than a difference between a sampling time and a transmission time of a packet at the server so as to allow the server to carry out said adapting.

14. (original) The multimedia streaming system of claim 11, wherein said at least one parameter comprises a number specifying a maximum difference between the number of bytes that has been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting.

15. (original) The multimedia streaming system of claim 11, wherein the server comprises an adapting mechanism for adapting a sampling rate to the transmission rate based on said at least one parameter.

16. (original) The multimedia streaming system of claim 11, wherein said at least one parameter comprises a clock shift amount for preventing playout disruption in the client.

17. (original) The multimedia streaming system of claim 11, wherein the server comprises an adapting mechanism for adjusting a transmission rate.

18. (original) The multimedia streaming system of claim 11; wherein the server comprises an adapting mechanism for adjusting a sampling rate.

19. (original) The multimedia streaming system of claim 11, wherein the server comprises an adapting mechanism for adjusting both a transmission rate and a sampling rate.

20. (original) The multimedia streaming system of claim 11, wherein the server comprises a software program having at least a code for carrying out said adapting.

21. (currently amended) A software application product embodied in a computer readable storage medium having a software application for use in a client in a multimedia streaming network for adaptively controlling level of a receiver buffer in the client, the multimedia streaming network comprising a server capable for providing streaming data to the client, wherein the receiver buffer is used to compensate for difference between data transmission amount by the server and data usage amount by the client so as to allow the client to have sufficient amount of streaming data to play-out in a non-disruptive manner, said software ~~product~~ application comprising:

a code for defining at least one parameter that determines a rate adaptation operating range in the server so as to carry out rate adaptation between the server and the client; and

a code for adjusting a packet transfer delay variation based on said adapting.

22. (currently amended) The software application product of claim 21, wherein said at least one parameter comprises a minimum shift amount indicative of a difference between a sampling time and a transmission time of a packet at the server so as to allow at the server to carry out said rate adaptation.

23. (currently amended) The software application product of claim 21, wherein said at least one parameter comprises a target shift amount indicative of a shift amount greater than a difference between a sampling time and a transmission time of a packet at the server so as to allow the server to carry out said rate adaptation.

24. (currently amended) The software application product of claim 21, wherein said at least one parameter comprises a number specifying a maximum difference between the number of bytes that have been sent and the number of bytes that have been sampled so as to allow the server to carry out said rate adaptation.

25. (currently amended) The software application product of claim 21, wherein said at least one parameter comprises a clock shift amount for preventing playout disruption in the client.

26. (original) A terminal in a multimedia streaming network having at least a server for providing streaming data to the terminal, the terminal having a receiver buffer to compensate for difference between data transmission amount by the server and data usage amount by the terminal so as to allow the terminal to have sufficient amount of streaming data to play-out in a non-disruptive manner, said terminal comprising:

a mechanism for defining at least one parameter that determines a rate adaptation operating range in the server so as to allow the server to adapt the data transmission amount to a reception rate based on said at least one parameter; and

a mechanism for adjusting a packet transfer delay variation based on said adapting.

27. (original) The terminal of claim 26, wherein said defining mechanism comprises a software program having at least a code for defining said at least one parameter.

28. (original) The terminal of claim 26, wherein said adjusting mechanism comprises a software program having at least a code for adjusting the packet transfer delay variation.

29. (original) The terminal of claim 26, wherein said at least one parameter comprises a minimum shift amount indicative of a difference between a sampling time and a transmission time of a packet at the server so as to allow the server to carry out said adapting based on the minimum shift amount.

30. (original) The terminal of claim 26, wherein said at least one parameter comprises a target shift amount indicative of a shift amount greater than a difference between a sampling time and a

transmission time of a packet at the server so as to allow the server to carry out said adapting based on the target shift amount.

31. (original) The terminal of claim 26, wherein said at least one parameter comprises a number specifying a maximum difference between the number of bytes that have been sent and the number of bytes that have been sampled so as to allow the server to carry out said adapting based on the number.

32. (original) A network element in the multimedia streaming network having a least a terminal that receives streaming data from the network element, the terminal having a receiver buffer to compensate for difference between data transmission amount by the network element and data usage amount by the terminal so as to allow the terminal to have sufficient amount of streaming data to play-out in a non-disruptive manner, said network element comprising:

means for receiving a request from the terminal, the request indicative of at least one parameter that determines a rate adaptation operating range in the network element; and

a mechanism for adapting, based on said at least one parameter, the data transmission amount to a reception rate by the terminal, so as to allow the terminal to adjust a packet transfer delay variation based on said adapting.

33. (original) The network element of claim 32, wherein said adapting mechanism comprises a software program having at least a code for adapting the data transmission amount.

34. (original) The network element of claim 33, wherein the software program comprises a code for adjusting the transmission rate.

35. (original) The network element of claim 33, wherein the software program comprises a code for adjusting a sampling rate.

36. (original) The network element of claim 33, wherein the software program comprises a code for adjusting of both a transmission rate and a sampling rate.